List of Claims:

Claim 1 (previously presented): A speech coding system comprising:

a preprocessor configured to receive a digitized signal from an analog-to-digital converter in time-domain, the preprocessor configured to transform the digital signal into frequency-domain, modify spectral magnitudes of the digitized signal in frequency-domain to generate a noise-reduced digitized signal and transform the noise-reduced digitized signal back to time-domain;

an encoder disposed to receive the noise-reduced digitized signal in time-domain, the encoder to provide a bitstream based upon a speech coding of the noise-reduced digitized signal;

where the speech coding determines at least one gain scaling a portion of the noise-reduced digitized signal; and

where the encoder adjusts the at least one gain as a function of noise characteristic for attenuating background noise in at least one frame,

wherein the at least one gain is adjusted according to a gain factor, the gain factor facilitating time-domain background noise attenuation.

Claim 2 (previously presented): The system according to Claim 1, where the speech coding comprises code excited linear prediction (CELP).

Claim 3 (previously presented): The system according to Claim 1, where the speech coding comprises extended code excited linear prediction (eX-CELP).

Claim 4 (previously presented): The system according to Claim 1, where the at least one gain is adjusted prior to quantization by the speech coding.

Claim 5 (previously presented): The system according to Claim 1, where the encoder adjusts the at least one gain according to the gain factor.

Claim 6 (previously presented): The system according to Claim 5, where the gain factor Gf is determined by the equation,

$$Gf = 1 - C \cdot NSR$$

where NSR has a value of about 1 when the portion comprises essentially background noise, where NSR is the square root of background noise energy divided by signal energy when the portion comprises speech, and where C is in the range of 0 through 1.

Claim 7 (previously presented): The system according to Claim 6, where C is in the range of about 0.4 through about 0.6.

Claim 8 (previously presented): The system according to Claim 6, further comprising a voice activity detector (VAD) operatively connected to the encoder, the VAD to determine when the portion comprises speech.

Claim 9 (previously presented): The system according to Claim 5, where the gain factor is based on a running mean.

Claim 10 (previously presented): The system according to Claim 9, where the running mean Gf new is determined by the equation,

$$Gf_{new} = \alpha \cdot Gf_{old} + (1 - \alpha) \cdot Gf_{current}$$

where Gf_{old} is a preceding gain factor for a preceding portion of the digitized signal, where $Gf_{current}$ is the gain factor based on the portion of the digitized signal, and where $0 \leq \alpha < 1$.

Claim 11 (previously presented): The system according to Claim 10, where α is equal to about 0.5.

Claim 12 (previously presented): The system according to Claim 1, where the portion of the digitized signal is one of a frame, a sub-frame, and a half frame.

Claim 13 (previously presented): The system according to Claim 1, where the encoder comprises a digital signal processing (DSP) chip.

Claim 14 (cancelled):

Claim 15 (previously presented): The system according to Claim 1, further comprising a decoder operatively connected to receive the bitstream from the encoder, the decoder to provide a reconstructed signal based upon the bitstream.

Claim 16 (previously presented): A speech coding system comprising:

a decoder disposed to receive a bitstream, the decoder to provide a reconstructed signal based upon a speech decoding of the bitstream;

where the speech decoding determines at least one gain scaling a portion of the reconstructed signal wherein the decoder adjusts the at least one gain as a function of noise characteristic for attenuating background noise in at least one frame and generating a background noise attenuated signal, and wherein the at least one gain is adjusted according to a gain factor, the gain factor facilitating time-domain background noise attenuation; and

a postprocessor configured to receive the background noise attenuated signal in time-domain, the postprocessor configured to transform the background noise attenuated signal into frequency-domain, modify spectral magnitudes of the background noise attenuated signal in frequency-domain to generate a noise-reduced attenuated signal and transform the noise-reduced attenuated signal back to time-domain.

Claim 17 (previously presented): The system according to Claim 16, where the speech decoding comprises code excited linear prediction (CELP).

Claim 18 (previously presented): The system according to Claim 16, where the speech decoding comprises extended code excited linear prediction (eX-CELP).

Claim 19 (previously presented): The system according to Claim 16, where the at least one gain is adjusted after decoding by the speech decoding.

Claim 20 (previously presented): The system according to Claim 16, where the decoder adjusts the at least one gain according to the gain factor.

Claim 21 (previously presented): The system according to Claim 20, where the gain factor Gf is determined by the equation,

$$Gf = 1 - C \cdot NSR$$

where NSR has a value of about 1 when the portion comprises essentially background noise, where NSR is the square root of background noise energy divided by signal energy when the portion comprises speech, and where C is in the range of 0 through 1.

Claim 22 (previously presented): The system according to Claim 21, where C is in the range of about 0.4 through about 0.6.

Claim 23 (previously presented): The system according to Claim 21, further comprising a voice activity detector (VAD) operatively connected to the decoder, the VAD to determine when the portion comprises speech.

Claim 24 (previously presented): The system according to Claim 20, where the gain factor is based on a running mean.

Claim 25 (previously presented): The system according to Claim 24, where the running mean Gf_{new} is determined by the equation,

$$G_{new} = \alpha \cdot Gf_{old} + (1 - \alpha) \cdot Gf_{current}$$

where Gf_{old} is a preceding gain factor for a preceding portion of the reconstructed signal, where $Gf_{current}$ is the gain factor based on the portion of the reconstructed signal, and where $0 \leq \alpha < 1$.

Claim 26 (previously presented): The system according to Claim 25, where α is equal to about 0.5.

Claim 27 (previously presented): The system according to Claim 16, where the portion of the reconstructed signal is one of a frame, a sub-frame, and a half frame.

Claim 28 (previously presented): The system according to Claim 16, where the decoder comprises a digital signal processing (DSP) chip.

Claim 29 (previously presented): The system according to Claim 16, further comprising an encoder operatively connected to provide the bitstream to the decoder.

Claim 30 (previously presented): A speech coding system comprising:

a preprocessor configured to receive a digitized signal from an analog-to-digital converter in time-domain, the preprocessor configured to transform the digital signal into frequency-domain, modify spectral magnitudes of the digitized signal in frequency-domain to generate a noise-reduced digitized signal and transform the noise-reduced digitized signal back to time-domain;

an encoder disposed to receive a the noise-reduced digitized signal, the encoder to provide a bitstream based upon a speech coding of the noise-reduced digitized signal, where the speech coding determines at least one gain scaling a portion of the digitized signal, and where the encoder adjusts the at least one gain as a function of noise characteristic; and

a decoder operatively connected to receive the bitstream from the encoder, where the decoder provides a reconstructed signal based upon a speech decoding of the bitstream, where the speech decoding reconstructs the at least one gain scaling the portion of the digitized signal, and where the decoder adjusts the at least one gain as a function of noise characteristic for attenuating background noise in at least one frame and generating a background noise attenuated signal, wherein the at least one gain is adjusted according to a gain factor, the gain factor facilitating time-domain background noise attenuation.

Claim 31 (previously presented): The system according to Claim 30, where the speech coding and the speech decoding comprise code excited linear prediction (CELP).

Claim 32 (previously presented): The system according to Claim 30, where the speech coding and the speech decoding comprise extended code excited linear prediction (eX -CELP).

Claim 33 (previously presented): The system according to Claim 30, where at least one of the encoder and the decoder adjusts the at least one gain.

Claim 34 (previously presented): The system according to Claim 30, where at least one of the encoder and the decoder adjusts the gain according to a gain factor.

Claim 35 (previously presented): The system according to Claim 34, where the gain factor Gf is determined by the equation,

$$Gf = 1 - C \cdot NSR$$

where NSR has a value of about 1 when the portion comprises essentially background noise, where NSR is the square root of background noise energy divided by signal energy when the portion comprises speech, and where C is in the range of 0 through 1.

Claim 36 (previously presented): The system according to Claim 35, where C is in the range of about 0.4 through about 0.6 when one of the encoder and the decoder adjusts the gain by the gain factor.

Claim 37 (previously presented): The system according to Claim 35, where C is in the range of about 0.2 through about 0.4 when the encoder and the decoder adjust the gain by the gain factor.

Claim 38 (previously presented): The system according to Claim 35, further comprising a voice activity detector (VAD) operatively connected to at least one of the encoder and the decoder, the VAD to determine when the portion comprises speech.

Claim 39 (previously presented): The system according to Claim 34, where the gain factor is based on a running mean.

Claim 40 (previously presented): The system according to Claim 39, where the running mean Gf_{new} is determined by the equation,

$$Gf_{new} = \alpha \cdot Gf_{old} + (1 - \alpha) \cdot Gf_{current}$$

where Gf_{old} is a preceding gain factor for a preceding portion of the digitized signal, where $Gf_{current}$ is the gain factor based on the portion of the digitized signal, and where $0 \leq \alpha < 0$.

Claim 41 (previously presented): The system according to Claim 40, where α is equal to about 0.5.

Claim 42 (previously presented): The system according to Claim 30, where the portion of the digitized signal is one of a frame, a sub-frame, and a half frame.

Claim 43 (previously presented): The system according to Claim 30, further comprising:

a postprocessor configured to receive the background noise attenuated signal in time-domain, the postprocessor configured to transform the background noise attenuated signal into frequency-domain, modify spectral magnitudes of the background noise attenuated signal in frequency-domain to generate a noise-reduced attenuated signal and transform the noise-reduced attenuated signal back to time-domain.

Claim 44 (previously presented): The system according to Claim 30, where at least one of the encoder and the decoder comprises a digital signal processing (DSP) chip.

Claim 45 (previously presented): A method of speech coding comprising:

receiving a digitized signal in time-domain;

transforming the digital signal into frequency-domain;

modifying spectral magnitudes of the digitized signal in frequency-domain to generate a noise-reduced digitized signal;

transforming the noise-reduced digitized signal back to time-domain;

segmenting the noise-reduced digitized signal into at least one portion;

determining at least one gain scaling the noise-reduced digitized signal within the one portion;

adjusting the at least one gain as a function of noise characteristic; and

quantizing the at least one gain into a group of at least one bit for a bitstream,

where the at least one gain is adjusted as a function of noise characteristic for attenuating background noise in at least one frame,

wherein the at least one gain is adjusted according to a gain factor, the gain factor facilitating time-domain background noise attenuation.

Claim 46 (previously presented): The method of Claim 45, where the speech coding system comprises code excited linear prediction (CELP).

Claim 47 (previously presented): The method of Claim 45, where the speech coding system comprises extended code excited linear prediction (eX-CELP).

Claim 48 (cancelled)

Claim 49 (previously presented): The method of Claim 45, where the adjusting further comprises adjusting the at least one gain according to the gain factor.

Claim 50 (previously presented): The method of Claim 49, where the gain factor Gf is determined by the equation

$$Gf = 1 - C \cdot NSR$$

where NSR has a value of about 1 when the portion comprises essentially background noise, where NSR is the square root of background noise energy divided by signal energy when the portion comprises speech, and where C is in the range of 0 through 1.

Claim 51 (previously presented): The method of Claim 49, where the gain factor is based on a running mean.

Claim 52 (previously presented): The method of Claim 51, where the running mean Gf_{new} is determined by the equation,

$$Gf_{new} = \alpha \cdot Gf_{old} + (1 - \alpha) \cdot Gf_{current}$$

where Gf_{old} is a preceding gain factor for a preceding portion of the digitized signal, where

$Gf_{current}$ is the gain factor based on the portion of the digitized signal, and where $0 \leq \alpha < 1$.

Claim 53 (previously presented): The method of Claim 52, where α is equal to about 0.5.

Claim 54 (previously presented): The method of Claim 45, where the portion is one of a frame, a sub-frame, and a half frame.

Claim 55 (previously presented): A method of speech coding comprising:

decoding at least one gain from a group of at least one bit in a bitstream;

adjusting the at least one gain as a function of noise characteristic;

assembling the at least one gain into a portion of a reconstructed speech signal, where the at least one gain is adjusted as a function of noise characteristic for attenuating background noise in at least one frame and generating a background noise attenuated signal in time-domain, and wherein the at least one gain is adjusted according to a gain factor, the gain factor facilitating time-domain background noise attenuation;

transforming the background noise attenuated signal into frequency-domain;

modifying spectral magnitudes of the background noise attenuated signal in frequency-domain to generate a noise-reduced attenuated signal; and

transforming the noise-reduced attenuated signal back to time-domain.

Claim 56 (previously presented): The method of Claim 55, where the speech coding system comprises code excited linear prediction (CELP).

Claim 57 (previously presented): The method of Claim 55, where the speech coding system comprises extended code excited linear prediction (eX-CELP).

Claim 58 (previously presented): The method of Claim 55, where the adjusting further comprises adjusting the at least one gain according to the gain factor.

Claim 59 (previously presented): The method of Claim 58, where the gain factor Gf is determined by the equation

$$Gf = 1 - C \cdot NSR$$

where NSR has a value of about I when the portion comprises essentially background noise, where NSR is the square root of background noise energy divided by signal energy when the portion comprises speech, and where C is in the range of 0 through 1.

Claim 60 (previously presented): The method of Claim 58, where the gain factor is based on a running mean.

Claim 61 (previously presented): The method of Claim 60, where the running mean Gf_{new} is determined by the equation,

$$Gf_{new} = \alpha \cdot Gf_{old} + (1 - \alpha) \cdot Gf_{current}$$

where Gf_{old} is a preceding gain factor for a preceding portion of the digitized signal, where $Gf_{current}$ is the gain factor based on the portion of the digitized signal, and where $0 \leq \alpha < 1$.

Claim 62 (previously presented): The method of Claim 61, where α is equal to about 0.5.

Claim 63 (previously presented): A method of speech coding comprising:

receiving a digitized signal in time-domain;

transforming the digital signal into frequency-domain;

modifying spectral magnitudes of the digitized signal in frequency-domain to generate a noise-reduced digitized signal;

transforming the noise-reduced digitized signal back to time-domain;

segmenting ~~a~~ the noise-reduced digitized signal into at least one portion;

determining at least one gain representing the noise-reduced digitized signal within the one portion;

pre-adjusting the at least one gain as a function of noise characteristic;

quantizing the at least one gain into a group of at least one bit for a bitstream.

decoding the at least one gain from the group of at least one bit in the bitstream;

post-adjusting the at least one gain as a function of noise characteristic; and

assembling the at least one gain into a reconstructed speech signal,

where the at least one gain is adjusted as a function of noise characteristic for attenuating background noise in at least one frame,

wherein the at least one gain is adjusted according to a gain factor, the gain factor facilitating time-domain background noise attenuation.

Claim 64 (previously presented): The method of Claim 63, where the speech coding system comprises code excited linear prediction (CELP).

Claim 65 (previously presented): The method of Claim 63, where the speech coding system comprises extended code excited linear prediction (eX-CELP).

Claim 66 (previously presented): The method of Claim 63, where at least one of the pre-adjusting and the post-adjusting further comprises adjusting the at least one gain according to the gain factor.

Claim 67 (previously presented): The method of Claim 66, where the gain factor Gf is determined by the equation

$$Gf = 1 - C \cdot NSR$$

where NSR has a value of about 1 when the portion comprises essentially background noise, where NSR is the square root of background noise energy divided by signal energy when the portion comprises speech, and where C is in the range of 0 through 1.

Claim 68 (previously presented): The method of Claim 66, where the gain factor is based on a running mean.

Claim 69 (previously presented): The method of Claim 68, where the running mean Gf $_{new}$ is determined by the equation,

$$Gf_{new} = \alpha \cdot Gf\,old + (1 - \alpha) \cdot Gf_{current}$$

where Gf_{old} is a preceding gain factor for a preceding portion of the digitized signal, where $Gf_{current}$ is the gain factor based on the portion of the digitized signal, and where $0 \leq \alpha < 0$.

Claim 70 (previously presented): The method of Claim 69, where α is equal to about 0.5.